

Field of Invention

The present invention relates to hand held data collector and analyzer systems and particularly relates to a vibration data collector and analyzer system having improved frequency analysis.

Background and Summary of Invention

Hand held data collector and analyzer systems are typically used to collect vibration data from machines for use in predicting maintenance requirements. For example, a typical data collector and analyzer instrument may be programmed to receive a route from a host computer, and such route would include a list of machines, test points, and a set-up condition for each test point. There may be thirty machines in the route with ten test points on each machine, and for each test point there may be specified a frequency range to be analyzed, a type of analysis to be performed, a particular type or set of data to be stored, and similar other parameters. In response to commands from the user, the hand held instrument prompts the user as to the identity of the machine and the test point to be monitored, and it automatically sets up the instrument, for example, to accept the specified frequency range for the test point, perform the specified analysis and store the specified type or set of data. A Fast Fourier Transform analysis may be performed on a pre-selected frequency range of the data and all or part of the resulting frequency spectrum may be stored and displayed. As the user progresses through the thirty machines and the corresponding 300 test points, he collects and stores vibration data which is subsequently transferred to the host computer for long term storage and further analysis.

Such hand held instruments are subject to rather severe weight and size constraints imposed by the need to be hand held and also must operate in a very hostile environment, such as a power plant or a heavy industrial manufacturing facility. In this environment, the instruments are exposed to a wide variety of vibration as well as a wide variety of temperatures, pressures, gas, dust and other atmospheric conditions. In such environment, the hand held instruments must be able to monitor vibration accurately in the presence of extreme noise. For example, one may wish to monitor a particular test point for a

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possible crack in a shaft bearing, where the bearing is rotating at 200 hertz, or 12,000 RPM, for example, and the bearing has twenty ball bearings. A crack in such bearing would produce a relatively high frequency, low amplitude, vibration or click each time a ball bearing passed over the crack. If such bearing is located in a typical factory, it will be operating in the presence of noise vibration from many sources such as stamps, presses, roller mills, conveyors, pumps, motors, etc.

Thus, to distinguish a click associated with a crack in a bearing, careful analysis of the frequency content of the monitor vibration signal is performed. Preferably, a user would select a frequency range of interest and look for a specific frequency that indicates or may indicate a cracked bearing. Typical known data collector and analyzer systems use analog filters to remove some of the unwanted frequency components. Usually a frequency spectrum is generated in a specific range and then specific data from the spectrum is inspected, compared or just stored.

One approach to providing the frequency analysis capability has been to provide a variable low-pass analog filter whose cutoff frequency can be controlled, and the output of the filter is sampled to provide a digital signal that is operated on by a computer to produce a frequency spectrum. This approach is generally acceptable, but it is not without limitations.

For example, analog low-pass filters will produce a certain amount of distortion of amplitude, frequency and phase, and variable low-pass filters tend to have greater distortion than fixed low-pass filters, particularly at or around the upper cutoff frequency of the filter. Such distortion is acceptable for many applications, but it certainly is not a desirable characteristic. In certain sensitive applications, such distortion may disqualify an instrument.

In addition to limitations at the higher end of the frequency range, some prior art instruments are known to be inaccurate and even unstable at low frequencies, particularly at frequencies on the order of one hertz or less. This characteristic usually results from circuit designs focused on higher frequencies because such frequencies are generally of greater interest. Again, however, in some applications accurate monitoring of low frequency signals is critical.

Another frequency related problem is the manner in which

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2 hand held data ~~collector~~ <sup>storage</sup> and analyzer instruments produce output  
that is limited to a frequency band of interest. One approach  
might be to use variable filters, such as switched capacitor  
filters, for which both a low-pass and a high-pass cutoff may be  
5 specified, or banks of filters may be used. Generally, this  
approach is not practical for a hand held device.

Another approach has been to provide a variable low-pass  
filter whose output is converted to a digital signal, and a Fast  
Fourier Transform is performed on the digital signal to produce a  
10 spectrum. Then, only the frequency band of interest in the  
spectrum is displayed or stored. This technique can be wasteful  
in terms of time, processor demands and accuracy. For example,  
to provide a 100 line spectrum between 9 kilohertz and 10  
15 kilohertz, such an approach calculates a 1,000 line spectrum  
between 0 and 10 kilohertz, essentially discards the lower 900  
lines, and displays the upper 100 lines. The discarded 900 lines  
represent waste.

These and other problems are addressed by the present  
invention which uses a fixed filter design to overcome the  
20 problems associated with variable filter designs. The computer  
source code for the present invention is listed in its entirety  
in the appendix. In accordance with the present invention, a  
hand held vibration data collector and analyzer system includes a  
vibration transducer producing an analog vibration signal. An  
25 input signal conditioning circuit receives and conditions the  
analog vibration signal from the vibration transducer to produce  
a conditioned analog signal, and it also includes a fixed  
frequency low-pass, or anti-aliasing, filter having an upper  
cutoff frequency set at a desired frequency for producing the  
30 conditioned analog signal with a desired frequency range. The  
input signal conditioning circuit further includes amplifiers for  
producing the conditioned analog signal at a desired amplitude, a  
D.C. offset circuit for removing D.C. components from the analog  
signal and an analog to digital converter for receiving and  
35 sampling the conditioned analog signal to produce a digital  
signal which is transmitted to a data processor for producing  
desired digital data.

The processor includes a transformer for selectively  
operating on the digital signal, such as performing a Fast  
40 Fourier Transform, and producing a frequency spectrum from the

2 digital signal. ~~The~~ The data processor also includes a selector for selecting and producing select data for storage from the digital signal or the frequency spectrum. A keyboard is interfaced with the data processor for inputting commands and data, and a display 5 is interfaced with the data processor for displaying information. A memory, interfaced with the data processor, stores information including the select data and means are provided for transferring information stored in the memory to another computer.

One advantage of the system described above is that the 10 fixed filter does not produce the same amount of distortion that is normally associated with variable low-pass filters, especially in the vicinity of the upper cutoff frequency of the filters. In addition, a selected frequency range of interest will normally be distant from the cutoff frequency of the fixed frequency low-pass 15 filter, which is not the case in designs using variable frequency low-pass filters, as discussed above. Thus, with the design in the present invention, the distortion in the vicinity of the cutoff frequency of the fixed frequency low-pass filter will normally not have an impact on the operation of the instrument.

20 In a preferred embodiment, the input conditioning circuit includes hardware for receiving and sampling the conditioned analog signal at a sample frequency that is substantially greater than the maximum frequency of interest to produce a digital signal and further, for digitally low-pass filtering and 25 digitally decimating the digital signal to produce the digital signal having a reduced sample rate and a selected upper cutoff frequency. In this embodiment, the data processor also includes a digital filter and decimator for optionally and selectively reducing the sample rate and frequency content of the digital 30 signal to produce a modified digital signal. Thus, in this particular embodiment, two separate portions of the system are designed to perform digital low-pass filtering and digital decimation.

In general, hardware performing digital filtering and 35 decimation is fast and precise, but generally, the variability of the output sample rates is limited. Digital low-pass filtering and decimation in the data processor is typically slower, but one may precisely choose the output sample rate and even the level of precision desired. By dividing the duty of digital filtering and 40 decimation between hardware in the input signal conditioning

circuit and the data processor, one may minimize the demands on the data processor without sacrificing precision, speed or variability. Also, by dividing this digital function between two portions of the system, one may efficiently impose other operations between the two digital filtering and decimation operations, again, without placing an undue or unrealistic demand on the data processor.

For example, in a preferred embodiment so-called true zoom processing is facilitated by the two digital filtering and decimation operations. In this embodiment, the data processor again includes a digital filter and decimator as described above. The data processor further includes a zoom processor for selectively and optionally operating on the digital signal from the conditioning circuit. To begin the zoom processing, the hardware of the conditioning circuit is actuated to digitally filter and decimate the input digital signal to produce a digital signal at a sample rate and with a frequency content as low as possible, but still greater than an upper frequency, which is the maximum frequency of interest in a frequency band of interest.

The zoom processor operates on the digital signal by frequency shifting the digital signal and at least low-pass filtering the frequency shifted digital signal to produce a zoom digital signal. The zoom digital signal is a selected band of frequencies in the digital signal ranging from a selected upper frequency, the maximum frequency of interest, to a selected lower frequency, and having a center frequency. Preferably, the zoom processor multiplies the digital signal by a function equal to  $e^{2\pi i(F_0)/(F_s)}$ , where  $F_0$  equals the center frequency of the selected band and  $F_s$  equals the sample rate of the digital signal.

This multiplication will effect a frequency shift of the digital signal such that the center frequency is shifted to zero. Then, preferably, the frequency shifted signal is subjected to low-pass filtering where the upper cutoff frequency of the low-pass filter is equal to:  $(f_u - f_b)/2$  where  $f_u$  = the upper frequency and  $f_b$  = the bottom or lower frequency in the frequency band of interest. A Fast Fourier Transform operation is performed on the low-pass filtered, decimated, and frequency shifted signal to produce a frequency spectrum that is then shifted up by  $f_b$ , the lower frequency, to become a frequency spectrum from  $f_b$  to  $f_u$ . By shifting the frequency of the digital

signal down before performing the filtering and transform operations, the speed and efficiency of the zoom processing is dramatically increased without any loss of accuracy.

To accommodate and process high frequency signals produced by the preferred embodiment of the present invention, the digital signal processor includes a digital signal processor (DSP) connected serially to receive the digital signal from the conditioning circuit for operating on the digital signal and, at least, for independently performing Fast Fourier Transforms to produce a frequency spectrum from the digital signal. The data processor further includes a central processing unit for controlling the operation of the system, including the operation of the DSP. In addition, a direct memory access (DMA) is provided in the CPU for transferring data directly to memory from the DSP to memory without interrupting the CPU.

In an alternate embodiment, the system includes a main conditioning circuit and an optional conditioning circuit, both of which include fixed frequency anti-aliasing filters, amplifiers and analog to digital converters. The main conditioning circuit includes a main microprocessor controller for controlling the main conditioning circuit, including the main analog to digital converter. The optional conditioning circuit does not include a separate microprocessor controller. Instead, control over the optional conditioning circuit is accomplished by and through an optional DSP.

Thus, this embodiment includes a main DSP connected to serially receive the main digital signal and an optional DSP connected to control said optional conditioning circuit and connected to serially receive the digital signal from the optional conditioning circuit. A CPU controls the overall operation of the system including issuing commands to the main and optional DSP's and the main microprocessor controller for controlling the optional conditioning circuit and the main conditioning circuit. In the preferred embodiment, the optional conditioning circuit is a separate optional circuit board and includes a plug system for connecting and disconnecting the optional circuit board to and from the system.

In the preferred embodiment, the memory for the system includes a main bank of random access memory (RAM) and a memory card that is selectively plugged into and out of communication

with the data processor. If a memory card is present and functioning properly, the data processor can store data directly to the memory card. However, if the memory card is missing, the data processor can store data in a pseudo-card that is configured in the RAM. In the preferred embodiment the user selects either the installed memory card, or the pseudo-card in RAM as the storage media. A pseudo-card in RAM is created by lower level programming and is designed to configure the memory in RAM in the same manner as a memory card. Thus, the upper level programming in the data processor stores data in only one format, namely the memory card format. To the data processor, the pseudo-card in RAM appears to be identical in data format to the actual memory card.

As previously mentioned, the problems of the prior art relating to frequency are not limited to high frequency problems. Inaccuracy and possible instability also appear in the very low frequencies. In the preferred embodiment, the input conditioning circuitry includes first and second analog to digital converters. The first analog to digital converter receives and samples the conditioned analog signal at a relatively high sample rate to produce a first digital signal, and the second analog to digital converter receives and samples the conditioned analog signal at a relatively low sample rate to produce a second digital signal. The second analog to digital converter includes precision quantizers that are accurate and stable in the conversion of low frequency signals in the range of about one hertz. A data processor receives and processes the first and second digital signals to produce select data for storage.

#### Brief Description of the Drawings

The present invention may best be understood by reference to a Detailed Description of preferred embodiments when considered in conjunction with the following Drawings, in which:

FIGURE 1 is a somewhat diagrammatical perspective view of a hand held vibration data collector and analyzer 10;

FIGURE 2 is a simplified block diagram of the hand held analyzer;

FIGURE 3 is a more detailed block diagram of the digital conditioning circuit of FIGURE 2;

FIGURE 4 is a somewhat more detailed block diagram of an analog to digital converter suitable for use in the circuit shown



in FIGURE 3;

FIGURE 5 is a block diagram showing an alternate embodiment for the analog to digital converter shown in FIGURE 3;

5 FIGURE 6 is a schematic block diagram of a preferred main digital board corresponding to the data processor, memory, display and keyboard of FIGURE 2;

FIGURE 7 is a schematic block diagram of a channel A board which, in a preferred embodiment, constitutes part of the digital conditioning block shown in FIGURE 2;

10 FIGURE 8 is a schematic block diagram of a channel B board which, in a preferred embodiment, constitutes part of the digital conditioning block shown in FIGURE 2;

Detailed Description

Referring now to the drawings in which like reference characters designate like or corresponding parts throughout the several views, there is shown in FIGURE 1 a hand held vibration data collector and analyzer 10. The height, width, and thickness dimensions of the analyzer 10 are 10.7" by 6.875" by 1.5" and it weighs approximately 4.25 pounds, including a battery. The analyzer 10 includes a keyboard 14 disposed on the face of the analyzer adjacent to a display 12. Control of the analyzer 10 is achieved through the keyboard 14 in combination with the display 12. In a typical operation, set-up conditions for a number of machines are downloaded into the analyzer 10 from a base computer through a connector 16 located at the top of the analyzer 10. Connector 16 is preferably an RS-232 compatible communications part that is also used to receive signals from transducers, such as transducer 18 (Fig. 2). A connector 17 provides a second input channel for receiving signals such as an RPM tach signal. Using the keyboard 14 and the display 12, the operator may cause the analyzer 10 to incrementally step through the downloaded route of machines, or may invoke independent analyzer functions that are totally independent of the route.

35 Referring to FIGURE 2, a simplified block diagram of the analyzer 10 is shown. A transducer 18, such as an acceleration, velocity, or displacement transducer, produces an analog signal, such as a vibration signal, that is supplied through line 19 to the analog conditioning circuit 20. Transducer 18 represents one or more transducers, such as an accelerometer and a tachometer, and connectors 16 and 17 are included in the analog conditioning

circuit 20. The term "line" as used herein should be regarded as both singular and plural. Depending on its context, "line" may refer to a single wire or electrical path, or it may collectively refer to a plurality of wires or electrical paths. Input control, analog filtering, analog integrating, amplification and D.C. offset functions are performed in the analog conditioning circuit 20 to produce a conditioned analog signal on line 22 which is supplied to a digital conditioning circuit 24. The conditioned analog signal is converted to a digital signal within digital conditioning circuit 24, and preferably, during the digital signal conditioning, the analog signal is Sigma Delta modulated, noise shaped, quantized to a digital signal, digitally filtered and digitally decimated. The conditioned digital signal is supplied through line 26 to a data processor 28, which is preferably a number of processors. The data processor 28 controls the digital conditioning circuitry through line 30 and controls the analog conditioning circuitry through line 32. Also connected to the data processor 28 are a memory 34, a display 12 and a keyboard 14. Preferably, the memory 34 includes RAM, ROM and a PCMCIA external memory card. The display 12 preferably is a luminous display manufactured by Samsung as model number 87-203-0123. The keyboard 14 is preferably a membrane matrix-type keyboard.

Also shown in FIGURE 2 is a battery 36 which powers all of the components shown in FIGURE 2. The battery 36 is physically connected to power all of the components shown in FIGURE 2.

Referring now to FIGURE 3, there is shown one possible embodiment of the analog conditioning circuit 20 and the digital conditioning circuit 24. As shown in FIGURE 3, the vibration signal is supplied by line 19 to an input control circuit 40. It will be understood that the input control 40 is connected to, and under the control of, the data processor 28. The operator may enter data through the keyboard 14 to instruct the data processor as to the type of transducer being used. Alternatively, the data processor will use downloaded data, or a default parameter, to determine the type of transducer being used. With such information, the data processor provides appropriate instructions to the input control 40 which configures the input for the desired transducer.

The output of the input control 40 is supplied to an

optional input filter 42 which is preferably a one kilohertz high-pass filter and may be used for envelope demodulation as described in more detail in U. S. Patent No. 5,249,138. The output of the filter 42 is supplied to integrators 44 which, 5 under the control of the data processor 28, integrate the input signal once, twice, or not at all. Thus, an acceleration signal may be converted to a velocity signal by integrating once; or used as an acceleration signal. Likewise, a velocity signal can be converted to a displacement signal by integrating once, and an 10 acceleration signal can be converted to a displacement signal by using two stages of integration.

The output from the integrators 44 is provided to a D.C. offset 46 that eliminates D.C. components from the signal, and a gain control 48 selects an appropriate gain and amplifies the 15 signal from the D.C. offset 46 to achieve desired amplitude levels in the signal.

The output from the gain control 48 is supplied to a fixed low-pass filter 50. Preferably, this low pass filter 50 is a six pole Butterworth low-pass filter having a frequency cut-off of 40 20 kilohertz. The output of the filter 50 is applied through line 22 to an analog to digital converter (ADC) 52 that converts the analog signal to a digital signal and supplies the digital signal to a digital filter 54. The digital filter 54 is under the 25 control of the data processor 28 and digitally filters the digital signal as indicated by the data processor 28. The output of the digital filter 54 is supplied to a digital decimator 56 whose function is to reduce the number of samples in the digital signal. The output of the decimator 56 is supplied on line 26 to the data processor. Again, it will be understood that the 30 components 40, 42, 44, 46, 48, 52, 54, and 56 are connected to and controlled by the data processor 28.

Referring to FIGURE 3, one design philosophy of the analyzer 35 10 may be understood. First, it should be noted that a single fixed frequency low-pass filter 50 is provided, operating at a cut-off frequency of 40 kilohertz. This fixed frequency low-pass filter 50 is in contrast to prior art hand held devices that used variable frequency analog filters that had variable cut-off frequencies, or larger prior art devices that use switchable banks of filters to achieve variable cut-off frequencies, all in 40 the analog domain. The ability to change the analog cut-off

frequency of the analog low-pass filters was considered important in prior art devices. However, analyzer 10 makes no attempt to change the analog cut-off frequencies and all low-pass filtering is conducted at a fixed 40 kilohertz rate. By using a fixed frequency low-pass filter, as opposed to variable frequency low-pass filters, all of the accuracy, fidelity and distortion problems associated with variable frequency filters are avoided.

Continuing with reference to FIGURE 3, it should be noted that the ADC 52 preferably operates at sampling frequencies of 3.276, 2.6218, 1.96608, 1.6384, 1.3107, 0.98304, 0.65536, and 0.3276, where all values are in megahertz. In the preferred embodiment the data processor 28 controls the clock frequency provided to the ADC 52. Since the low-pass filter 50 operates at 40 kilohertz, one would normally operate the ADC 52 at about 40 kilohertz in accordance with the Nyquist sampling theorem, in order to avoid aliasing problems for a maximum frequency of interest of about 20 kilohertz. However, operating at the sampling rate suggested by the Nyquist theorem would require an extremely sharp cut-off frequency to avoid distortion near the cut-off frequency. In other words, to achieve high fidelity, the cut-off of the low-pass filter must be almost vertical if the Nyquist sampling rate is used. However, by sampling at a much higher rate than necessary and then digitally filtering and decimating the signal, the output signal has the desired sampling frequency, and has increased resolution afforded by the decimation process. Additionally, the requirement for a vertical cut-off at the low-pass frequency by the anti-aliasing filter is not nearly as stringent as when sampling at lower frequencies, and this relaxed requirement introduces no associated reduction in digital signal quality.

In the preferred embodiment, the maximum frequency of interest is 20 kilohertz. According to the Nyquist theorem, the sampling frequency for an input signal of 20 kilohertz should be approximately 40 kilohertz. The cut-off frequency of the low-pass filter 50 is chosen so as to allow sampling of this 20 kilohertz maximum frequency. However, in the preferred embodiment, the chosen output sampling frequency of the filtered and decimated 20 kilohertz signal is 51.2 kilohertz, just somewhat above the minimum Nyquist frequency of 40 kilohertz.

Since the sampling frequency will be reduced by a factor of

64:1 through the filtering and decimation process, the ADC 52 must be operated at a frequency 64 times the desired output sampling frequency of 51.2 kilohertz. This requires the ADC 52 to operate at 3.276 megahertz, which requires a clock frequency of 6.5536 megahertz, which represents the maximum frequency at which the ADC 52 will need to operate, to produce a digital representation of the maximum input frequency of interest, 20 kilohertz, with a resultant sampled frequency of 51.2 kilohertz. If additional filtering and decimation are required, it can be accomplished in the data processor 28 as will be hereinafter described in greater detail.

In the preferred embodiment, the minimum frequency of interest is 2 kilohertz. According to the Nyquist theorem, the sampling frequency for an input signal of 2 kilohertz should be approximately 4 kilohertz. However, since a fixed frequency low-pass filter is being used, the cut-off frequency of the low-pass filter 50 must be chosen so as to allow sampling of the 20 kilohertz maximum frequency. Aliasing problems in the sampled signal would normally occur with a low-pass filter set at a frequency so high above the Nyquist frequency for the frequency of interest. However, because of the high rate of input frequency oversampling and subsequent decimation, aliasing of the 2 kilohertz signal, even when low-pass filtered above the Nyquist frequency, is not a problem. In the preferred embodiment, the chosen output sampling frequency of the filtered and decimated 2 kilohertz signal is 5.12 kilohertz, just somewhat above the minimum Nyquist frequency of 4 kilohertz.

Since the sampling frequency will be reduced by a factor of 64:1 through the filtering and decimation process, the ADC 52 must be operated at a frequency 64 times the desired output sampling frequency of 5.12 kilohertz. This requires the ADC 52 to operate at 327.6 kilohertz, which requires a clock frequency of 655.36 kilohertz, which represents the minimum frequency at which the ADC 52 will need to operate, to produce a digital representation of the minimum input frequency of interest, 2 kilohertz, with a resultant sampled frequency of 5.12 kilohertz. If additional filtering and decimation are required, it can be accomplished in the data processor 28 as will be hereinafter described in greater detail.

Referring now to FIGURE 4, there is shown a preferred

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embodiment of the ADC 52, filter 54, and decimator 56. Referring to FIGURE 4, the analog signal appears on line 22 and is received by a third order cascaded Sigma Delta loop 60 having a transfer function of  $Y(z) = X(z) + (1-z^{-1})^3 Q^3(z)$  where  $Y(z)$  is the digital output,  $X(z)$  is the analog input, and  $Q^3$  constitutes the quantization noise produced by the third cascaded Sigma Delta loop. A digital signal  $Y(z)$  is produced by the Sigma Delta loop 60 on line 62 and is directed to a comb filter 64. Preferably, the comb filter 64 filters and decimates the incoming signal at a ratio of 16:1. That is, the output sampling rate is reduced by a factor of 16 as compared to the input sampling rate. A finite impulse response (FIR) filter 66 receives the output of the comb filter 64 and provides a 4:1 reduction of the sampling rate.

In the preferred embodiment, the maximum output sampling frequency of the Sigma Delta loop 60 is 3.276 megahertz with one bit of resolution. The comb filter 64 receives the digital signal on line 62 and decimates the signal by a factor of 16. Thus, the maximum output of the comb filter is 204.8 kilohertz, but the signal now has 12 bits of resolution.

The FIR filter 66 receives the output of the comb filter 64 and decimates the signal by a factor of 4. Thus, the output of the FIR filter 66 is 51.2 kilohertz with 16 bits of resolution. It will be understood that the sampling rate of the Sigma Delta loop 60 and the resultant input and output rates of the comb filter 64 and FIR filter 66 are dependent upon a clock frequency, and can be reduced from the numbers previously described by reducing the input clock. In the preferred embodiment, the maximum input sampling rate is 3.276 megahertz, generated by the maximum internal input clock rate of 6.5536 megahertz. After a decimation of 64:1, this results in an output sampling rate at the FIR filter 66 of 51.2 kilohertz. However, the output sampling rates from the FIR filter 66 may be adjusted from 51.2 kilohertz down to 5.1 kilohertz by adjusting the input clock frequency down to 655.36 kilohertz. Likewise, the output sampling rates of the comb filter 64 may be adjusted from 204.8 kilohertz down to 20.48 kilohertz.

Referring now to FIGURE 5, an alternate embodiment of the ADC 52 is shown. In the embodiment of FIGURE 5, the ADC 52 is two converters, namely, a fast ADC 70 and a slow ADC 72. The slow ADC 72 operates at an input sampling rate of 20 hertz and

produces an output sample rate of 20 hertz with 12 bits of resolution. The fast ADC 70 operates as described for Sigma Delta loop 60, comb filter 64, and FIR filter 66 as depicted in FIGURE 4. The signals from both ADC 70 and 72 are provided to the data processor 28 on lines 26a and 26b, respectively. The data processor 28 may choose to process the signal of one or both of the ADC's 70 and 72 depending upon its operator instructions or a downloaded program. Preferably, the slow ADC is model number LTC1294, manufactured by Linear Technology Incorporated, and the fast ADC is model number DSP56ADC16, manufactured by Motorola.

Generally, a fast ADC operates at a fast input sampling rate which may produce accuracy or stability problems when receiving low frequency signals, especially those signals having a frequency on the order of 1 hertz or less. It is known that some prior art devices become unstable when input signals have a frequency of less than 1 hertz. Therefore, by the provision of a slow ADC 72, low frequency signals are sampled in a stable and accurate manner. In some applications, low frequency signals are extremely important and must be measured with great accuracy to achieve meaningful results. Thus, by the provision of a slow ADC 72, such measurements may be made without sacrificing the speed of the instrument and without accepting accuracy limitations of fast ADC's such as disclosed with reference to FIGURE 4.

Referring now to FIGURES 6, 7 and 8, a detailed disclosure of a preferred embodiment is shown. In this particular embodiment, the analyzer 10 utilizes a channel A board shown in FIGURE 7, which may be regarded as a permanent part of the analyzer 10, and an optional channel B board is shown in FIGURE 8. The channel B board of FIGURE 8 is designed with increased simplicity as compared to the channel A board of FIGURE 7, and detachably plugs into a connection with the channel A board of FIGURE 7.

Referring now to FIGURE 6, the main digital board includes a central processing unit (CPU) 80 which is preferably a Z8S180 manufactured by Zilog. The CPU 80 includes an internal DMA allowing other processors to directly access memory without interrupting the CPU 80. An EPROM 82 is interfaced with the CPU 80 and functions primarily to store the operating program of the CPU 80. Also, a system memory 83 is interfaced with the CPU 80,

and it stores data downloaded by computer, collected by the analyzer 10, or provided by the operator. It also stores the operating program when analyzer 10 is operating. A PCMCIA memory card 84 is connected through a conventional memory plug 85 to the central processing unit 80, and its primary function is to store collected data. By means of low level programming executed by the CPU 80, the RAM system memory 83 is configured to appear as a PCMCIA memory card, and this appearance will be referred to as pseudo-card memory. By employing pseudo-card memory, the CPU 80 utilizes the same format for storing data even though two types of memory are utilized. When the RAM system memory 83 is being utilized, it is utilized as a pseudo-card, and thus the format of the data supplied to or received from the RAM system memory 83 is identical to the format of the data supplied to or received from the PCMCIA memory card 84.

In addition to reducing memory complexity, the use of pseudo-card memory reduces the programming needed to store data in two types of memory, and it reduces the programming and processing overhead required when data is transferred from RAM system memory 83 to the PCMCIA memory card 84, or vice versa. Data that is collected by the CPU 80 and downloaded into the PCMCIA memory card 84 may be transferred through an RS232 level translation interface 86 to a base computer 88. It is understood that the RS232 interface 86 is part of the analyzer 10 and is connected and disconnected from the base computer 88 as desired.

As opposed to transferring data through the interface 86 to the base computer 88, one may also transfer data to the base computer 88 by unplugging the PCMCIA memory card 84 from plug 85 and plugging the card into the base computer 88. In such case, the base computer would simply access and read the PCMCIA memory card 84 directly to receive the collected and analyzed data.

A digital MPU 90 is also interfaced with CPU 80 for reading keyboard 14 and a serial EEPROM 92. In the preferred embodiment, the MPU 90 is a model 68HC705C8, manufactured by Motorola. The primary function of the MPU 90 is to receive signals from the keyboard 14 and interface such signals with the CPU 80 through appropriate interrupt programming. In addition, the MPU 90 is interfaced with serial EEPROM 92 which stores specialized data, such as a unique serial number to identify a particular analyzer 10. The downloaded serial number is particularly useful in

electronically identifying a particular analyzer 10 so that programs can independently determine whether they are loaded on the correct analyzer 10. This serial number may be regarded as a security device for both the owner of the analyzer 10 and the supplier of software for use in the analyzer 10.

A real time clock 94 is also interfaced with the CPU 80, and the CPU 80 provides clock signals to all digital components requiring such signals. An LCD display 96, preferably model number 87-203-0123, manufactured by Samsung is connected through an LCD controller 98 to the CPU 80. The LCD controller 98 is preferably model number E1330, manufactured by Epson, and it includes its own memory subsection to store at least one screen of information. The memory 34 as shown in FIGURE 2 may be regarded as the EPROM 82, the RAM system memory 83, the PCMCIA memory card 84, the memory of MPU 90, the EEPROM 92, the memory of LCD controller 98, and the DSP memory 182. The CPU 80 communicates with the channel A board of FIGURE 7 through lines 100, 102, and 104, and communicates with the channel B board of FIGURE 8 through line 106.

Referring now to FIGURE 7, line 100 interfaces CPU 80, depicted in FIGURE 6, with MPU 110, which is preferably model number 68HC705C8, manufactured by Motorola, and line 100 is preferably a digital bus. The MPU 110 functions to control the overall operation of the channel A board of FIGURE 7. Through line 112, the MPU 110 controls an input control 114 which corresponds to input control 40 shown in FIGURE 3. The input signal on line 19 may be an acceleration, velocity, or displacement signal or it could represent temperature, pressure or any other parameter that may be represented by an electrical signal. The input control 114 configures the analyzer 10 to appropriately receive the signal on input line 19.

The signal from the input control 114 is applied through line 116 to a 1 kilohertz high-pass filter 118, and a switch 120 that is controlled through lines 122 by the MPU 110. The output of the filter 118 is applied through line 124 to an integration circuit 126 whose operation is controlled through line 128 by the MPU 110. The integration circuit 126 will integrate the signal once or twice depending upon the control input from the analog MPU 110. The output of the filter 118 is also applied through line 124 to a switch 130 that also receives the output of the

switch 120. Switch 130 is controlled through line 122 by the MPU 110. The output of the integration circuit 126 and the output of the switch 130 are applied to a switch 132 which is controlled by the MPU 110 through line 122. Through switches 120, 130 and 132, it will be appreciated that the MPU 110 can bypass or use filter 118, and bypass or use the integration circuit 126 to produce an output signal at the output of switch 132.

An adder 134 receives the output of the switch 132 through line 136, and the other input of adder 132 is provided by a D.C. offset adjust 138 which is supplied through line 140. The D.C. offset adjust 138 is controlled by MPU 110 through line 142. The CPU 80 monitors the output signal of the channel A board, and through the MPU 110 controls the D.C. offset adjust 138 to eliminate D.C. components from the output of the channel A board. The output of adder 134 is applied through line 144 to analog gain stages 146 that are controlled by the MPU 110 through line 148. The analog gain stages 146, in conjunction with the MPU 110, perform an autoranging function so that the output of the stages 146 are supplied within a desired amplitude range. The output of the analog gain stages 146 is supplied through line 150 to a low-pass filter 160, which is preferably a fixed frequency, six pole, 40 kilohertz, Butterworth low-pass filter.

The output of the low-pass filter 160 is applied through line 174 to an ADC 176, preferably model DSP56ADC16, manufactured by Motorola, whose output is applied through line 178 to a digital signal processor (DSP) 180, preferably a DSP 56002, manufactured by Motorola. The output of the converter 156 is also supplied through line 154 to an RMS to DC converter 156. The output of the converter 156 is supplied through line 162 to a slow ADC 164, which is preferably model number LTC1294, manufactured by Linear Technology, Inc. The output of the ADC 164 is applied through line 172 to the MPU 110.

The output of the DSP 180 is supplied by lines 102 to the CPU 80. In the preferred embodiment, the output of the DSP 180 is supplied almost directly to the RAM system memory 83 through the DMA of the CPU 80 without interrupting the CPU 80. The DSP 180 both sends and receives signals from the CPU 80 through line 102. Thus, under the control and instruction of the CPU 80, in the preferred embodiment, the DSP 180 performs low-pass filtering operations, zoom operations, Fast Fourier Transform operations,

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and other manipulations of the incoming data, and then transfers the processed data to the RAM system memory 83 through the DMA without interrupting the CPU 80.

The DSP 180 is interfaced with its own program memory 182 through line 184. The sampling frequency of the ADC 176 is controlled by the sampling frequency select 186 through line 188. Select 186 is controlled by the CPU 80 through line 104.

Referring now to FIGURE 8, the channel B board is shown. The channel B board may be plugged into, or unplugged from, the analyzer 10 by a conventional digital plug system 200, which is preferably a model 532956-1 manufactured by AMP, Inc. The channel B board includes an analog processing portion 202 that is a simplified version of the analog section shown in FIGURE 7. The analog processing portion 202 includes an input control 204 analogous to input control 114 of FIGURE 7. Likewise, the integrator 206 is analogous to integration circuit 126. Finally, the gain circuit 210 is analogous to the analog gain stages 146.

The components 204, 206, and 210 of the analog processing portion 202 are all under the control of a DSP 212 mounted on the channel B board and connected to the analog processing portion 202 through line 214. Thus, the DSP 212 has additional control duties that are not assumed by the DSP 180 of the channel A board depicted in FIGURE 7. This structure facilitates the plug-in capability of the channel B board. In the channel A board, the control functions were provided by the MPU 110. The output of the analog processor portion 202 is supplied through line 216 to an ADC 218, which is preferably a model DSP56ADC16 manufactured by Motorola. The sampling frequency of the ADC 218 is controlled by the sampling frequency select 186, as seen in FIGURE 7, through line 220. As previously noted, the sampling frequency select 186 is controlled by the CPU 80, and thus both ADC's 176 and 218, when operating simultaneously, will operate at the same sampling frequency. The output of the ADC 218 is supplied through line 221 to the DSP 212 whose output is supplied through line 106 and plug 200 to the CPU 80. Also, the DSP 212 is connected by line 222 to a DSP program memory 224.

In a typical operation, the base computer 88 will download a route through the interface 86 to the CPU 80 which stores the route in RAM system memory 83. The route would typically include a list of machines, and for each list of machines a number of

test points would be specified. For each test point, set-up parameters and data storage parameters would be listed. The set-up parameters would include such things as the type of sensor to be used, the preferred units, the frequency range of interest, the type of data to be displayed, the specific data to be stored and other parameters associated with the measurement point, such as the identity of the machine, the identity of the measurement point, a description of the measurement point, the date and time, etc.

The CPU 80 downloads the parameters for one measurement point to the MPU 110 of the channel A board, and it downloads the parameters for a second measurement point to the DSP 212 of the channel B board. The CPU 80 also supplies the sampling frequency to the sampling frequency select 186, which is based upon the highest frequency of interest for the two test points being monitored by the channel A and B boards of FIGURES 7 and 8. In response to the sensor parameters, the MPU 110 properly configures the input control 114 and the DSP 212 configures the input control 204. Depending upon the type of input signal being received and the type of output signal that is desired, the MPU 110 will appropriately adjust and use or not use the integration circuit 126 and, likewise, the DSP 212 will adjust, use or not use the integrators 206.

Once the input signal is appropriately amplified and D.C. components are removed, the input signal is supplied through the ADC 176 to the DSP 180, and the other signal is supplied through the ADC 218 to the DSP 212. Each DSP 180, 212 then performs the desired function which was previously downloaded by the CPU 80. Typically, each DSP 180, 212 will perform a Fast Fourier Transform on the incoming signal to produce a frequency spectrum of interest. That frequency spectrum will typically be transferred directly to RAM system memory 83 through the CPU 80 and, likewise, the spectrum will be transferred from the RAM system memory 83 through the DMA of CPU 80 to the LCD controller 98 for display on the LCD display 96.

It will be appreciated that the DSP's 180 and 212 are serially receiving the digital input signal and are calculating and analyzing the data independently of the CPU 80, without interrupting the CPU 80. Specifically, both DSP's 180, 212 typically will perform Fast Fourier Transform calculations and

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then perform a rasterization function to graphically display the spectrum produced by the Fast Fourier Transform. Each DSP 180, 5 212 also identifies data that needs to be stored in accordance with instructions previously received from the CPU 80. For example, once a spectrum is calculated, the stored instructions may specify storage of the amplitude of the spectrum at six selected frequencies. In such case, the DSP's 180 and 212 identify the six selected frequencies of interest and transfer the selected data through the DMA of the CPU to the PCMCIA memory 10 card 84, if the user so desires, where the data is stored in a memory card format. Alternatively, if a PCMCIA memory card 84 is not plugged in, the data can be stored in RAM system memory 83 in a pseudo-card format.

It will be appreciated that the DSP's 180 and 212 operate 15 independently and serially upon the incoming signal to receive, analyze and store the data, all without interrupting the CPU 80. This leaves the CPU 80 free to conduct other control activities as previously explained and, therefore, greatly increases the speed of the overall analyzer 10 by distributing computation and 20 control responsibilities among DSP 180, DSP 212, CPU 80, MPU 110, LCD controller 98, and MPU 90. In particular, speed is enhanced by serially processing the data using the DSP's 180 and 212. It will be understood that when two channels are used, such as the channel A board of FIGURE 7 and the channel B board of FIGURE 8, 25 there are two input signals that are typically received from two different vibration transducers.

Since channel A and channel B function virtually independently, there is no need for the vibration sensors to be 30 of the same type, and in fact, the sensors may sense different parameters such as vibration and temperature. For purposes of simplicity, the design shown in FIGURES 6, 7 and 8 utilizes a frequency select 186 that locks the sampling frequencies of the two ADC's 176 and 218 such that each converter must have the same sampling frequency when operating simultaneously. This is a 35 design choice and it would be possible to operate the converters at different sampling rates if desired.

Referring again to FIGURE 3, one may best appreciate 40 advantages of the present invention. The advantages of a fixed low-pass filter have been previously discussed, but there are advantages in providing a digital filter 54 and a digital

5 decimator 56 as part of digital conditioning hardware positioned upstream from the data processor 28. For example, assume that an operator wanted to see a spectrum of the signal for frequencies between 9 and 10 kilohertz. Producing a frequency spectrum over this frequency band only is defined as a true zoom function. To begin the operation, the clock signal to the ADC 52, digital filter 54, and the digital decimator 56 is chosen so as to effectively low-pass filter the incoming signal at a frequency above 10 kilohertz but as close to 10 kilohertz as possible.

10 In the preferred embodiment, a clock signal of 3.276 megahertz would be chosen, and the effective low-pass cut-off frequency of the digital filter 54 would be about 10 kilohertz with an output sample rate of 25.6 kilohertz after filtering and decimation. The digital signal is then fed to the data processor  
15 28 where, under software processing, the incoming digital signal is frequency-shifted down by 9.5 kilohertz, which means that the center point, 9.5 kilohertz, of the frequency band of interest, 9 to 10 kilohertz, is shifted to 0 hertz. Thus, the frequency band of interest in the shifted signal now extends from negative 0.5  
20 kilohertz to positive 0.5 kilohertz. Next, the frequency shifted signal is digitally low-pass filtered in the data processor 28 using a cut-off frequency of 0.5 hertz. The digital low-pass filter removes frequency content in the signal that is above 0.5 kilohertz and is below negative 0.5 kilohertz. Thus, the  
25 remaining signal, negative 0.5 kilohertz to positive 0.5 kilohertz, constitutes the frequency band of interest.

30 A Fast Fourier Transform is then performed on this band of interest in the data processor 28 to produce a spectrum, and the spectrum is shifted up by 9.5 kilohertz to produce a frequency spectrum extending from 9 kilohertz to 10 kilohertz. That is, the frequency spectrum line at negative 0.5 kilohertz would be shifted up to 9 kilohertz, and the frequency spectrum line at 0.5 kilohertz would be shifted up to 10 kilohertz. All of the  
35 remaining frequency lines would be shifted up accordingly. The ability to first digitally filter and decimate the signal outside of the data processor 28 enables one to minimize the processing demands on the data processor 28, and thus enables this particular instrument to perform zoom operations on the fly in real time. Such zoom processing would be more difficult without  
40 the outside hardware performing filtering and decimation

functions. The zoom processor of the present invention may be best understood by reference to the source code in the appendix.

In accordance with another aspect of the invention, which has been previously described in some detail, the RAM system memory 83 is configured in a pseudo-card format. This format is achieved by using a pseudo-card low level program which is disclosed in the source code in the appendix.

Having described preferred embodiments of the invention, it will be appreciated that the vibration data analyzer achieves superior stability, accuracy and reliability through the use of fixed frequency anti-aliasing filters. In addition, stability and accuracy is further enhanced through the use of two ADC's, one for high frequency signals, and one for low frequency and D.C. signals. Efficient digital filtering and decimation is achieved by splitting the processing demands between specialized hardware filters and decimators and software filters and decimators implemented in a data processor. A true zoom method is employed to take advantage of the filtering and decimation capabilities of the analyzer, and frequency shifting techniques are used to reduce demands on the data processor when producing a true zoom.

By using independent DSP's that serially operate on incoming signals, the demands on the data processor, particularly the CPU, are further reduced. In addition, the RAM system memory is formatted as a pseudo-card to facilitate the use of a PCMCIA memory card or the RAM system memory. While preferred embodiments of the present invention have been described above, it will be appreciated by those of ordinary skill in the art that the invention is capable of numerous modifications, rearrangements and substitutions of parts without departing from the spirit of the invention.

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